CLAIMS

[1] A sampling rate converter comprising:

an up sampler for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[2] A sampling rate converter as set forth in claim 1, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by

performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[3] A sampling rate converter as set forth in claim 1, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter.

[4] A sampling rate converter as set forth in claim 1, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[5] A sampling rate converter as set forth in claim 1, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[6] A sampling rate converter comprising:

an up sampler for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes a filter coefficient, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[7] A sampling rate converter as set forth in claim 6, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

[8] A sampling rate converter as set forth in claim 6, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[9] A sampling rate converter as set forth in claim 6, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling

frequency of the input is higher than a sampling frequency of the output.

[10] A sampling rate converter comprising:

an up sampler for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[11] A sampling rate converter as set forth in claim 10, wherein the filter coefficient is set based on an

amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[12] A sampling rate converter as set forth in claim 10, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

[13] A sampling rate converter as set forth in claim 10, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[14] A sampling rate converter as set forth in claim 10, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[15] A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing the convolution processing of input sample signals and the poly-phase filters decomposed to the poly-phases,

a plurality of up samplers for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from the linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[16] A sampling rate converter as set forth in claim 15, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[17] A sampling rate converter as set forth in claim 15, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter.

[18] A sampling rate converter as set forth in claim 15, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[19] A sampling rate converter as set forth in claim 15, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[20] A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for inserting U-1 zero points between output signals of corresponding the convolution processing units and raising the sampling frequency U-fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and an impulse response becomes the filter coefficient, and

the filter coefficient is set by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[21] A sampling rate converter as set forth in claim 20, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

[22] A sampling rate converter as set forth in claim 20, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[23] A sampling rate converter as set forth in claim 20, further including a low bandpass filter preventing an imaging component from occurring and a non-original

frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[24] A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing

weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[25] A sampling rate converter as set forth in claim 24, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[26] A sampling rate converter as set forth in claim 24, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

[27] A sampling rate converter as set forth in claim 24, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[28] A sampling rate converter as set forth in claim 24, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[29] A sampling rate converter comprising:

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and

a linear interpolation block for finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[30] A sampling rate converter as set forth in claim 29, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency

response of the pre-filter.

[31] A sampling rate converter as set forth in claim 29, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter.

[32] A sampling rate converter as set forth in claim 29, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[33] A sampling rate converter as set forth in claim 29, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[34] A sampling rate converter as set forth in claim 29, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

[35] A sampling rate converter comprising:

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR

filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and

a linear interpolation block for finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[36] A sampling rate converter as set forth in claim 35, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[37] A sampling rate converter as set forth in claim 35, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is

lower than a sampling frequency of the output.

[38] A sampling rate converter as set forth in claim 35, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[39] A sampling rate converter as set forth in claim 35, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

[40] A sampling rate converter comprising:

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and

a linear interpolation block for finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse

response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[41] A sampling rate converter as set forth in claim 40, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[42] A sampling rate converter as set forth in claim 40, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

[43] A sampling rate converter as set forth in claim 40, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[44] A sampling rate converter as set forth in claim 40, further including a low bandpass filter preventing an

imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[45] A sampling rate converter as set forth in claim 40, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

[46] A sampling rate conversion method comprising:

a first step of inserting U-1 zero points between sample signals and raising the sampling frequency U-fold,

a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, wherein

the filter coefficient of the FIR filter is

calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[47] A sampling rate conversion method comprising:

a first step of inserting U-1 zero points between sample signals and raising the sampling frequency U-fold,

a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length and an impulse response becomes the filter coefficient, and

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, wherein

the filter coefficient of the FIR filter is calculated by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[48] A sampling rate conversion method comprising:

a first step of inserting U-1 zero points between sample signals and raising the sampling frequency U-fold,

a second step of performing predetermined

convolution processing with respect to a signal multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation, wherein

the filter coefficient of the FIR filter is calculated by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[49] A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals, and

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[50] A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals, and

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[51] A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised

U-fold and generating a signal obtained by adding all signals, and

a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency points to be passed and a frequency response of the pre-filter.

[52] A sampling rate conversion method comprising:

a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and

a second step of performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, wherein

the FIR filter is a FIR filter where an impulse

response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[53] A sampling rate conversion method comprising:

a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and

a second step of performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[54] A sampling rate conversion method comprising:

a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and

a second step of performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency points to be passed and a frequency response of the pre-filter.

[55] An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises

an up sampler for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from linear interpolation, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the pre-filter.

[56] An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing the weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the pre-filter.

[57] An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient and

a selector for selecting two points of samples required for an output sample and selecting the

coefficient of the corresponding poly-phase filter, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency to be passed and/or a frequency response of the pre-filter.